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relating to a

TELECOMMUNICATION SERVICES IDENTIFICATION IN A GATEWAY

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Telecommunication Services Identification

Field of the Invention

The present invention relates to a method and apparatus for identifying the type of telecommunications service required during call establishment over two networks. In particular, it relates to a method and apparatus for establishing call control appropriate to the telecommunications service, and more specifically to the service identified by the originating network.

Background of the Invention

Today, transmission of information via communications networks is commonplace. There is ever increasing demand for more rapid and flexible data exchange and the nature of the data to be transmitted is becoming ever more varied. Communications networks are also very diverse in scale and nature, from local area networks (LANs) comprising a handful of personal computers within a small company, to the Internet, providing the possibility for data exchange at a global level.

Communications networks have different characteristics and operating principles. Some are designed specifically for telecommunications and others are intended for more diverse data communications. Many different network architectures are employed, and a wide variety of data communication protocols are used to transmit information. Some networks, for example parts of the traditional Public Switched Telephone Network (PSTN) still utilise anologue data transmission techniques, while others, for example the Integrated Services Digital Network (ISDN) and the Public Land Mobile Network (PLMN), of which the GSM mobile telephone network is an example, employ digital communication technology.

Networks specifically designed for telecommunications still commonly use circuit switched data communication, in which specific network resources are allocated to establish and maintain a connection between end users of the network. Other networks, more typically, those designed primarily for general-purpose data communications, such as the Internet and local area networks, use packet switched communications protocols in which no dedicated physical connection must be established or maintained. In this kind of network, the continuous flow of data to be transmitted is divided into packets, each of which is routed, often in an independent connectionless manner, between the end users of the network.

In the traditional telecommunications sector, there is considerable interest in diversification and the provision of new telecommunications and datacomms services for network subscribers. In the data communications sector there is also increased interest in the provision of more telecomms oriented services. Thus, progressive overlap (or convergence) between the roles of telecommunications and data communications networks is occurring.

This convergence has given rise to a greater need to transfer data between dissimilar communications networks. However, because of the inherently diverse nature of currently existing telecommunications and data communications networks, a significant problem of inter-operability or interworking exists.

The problem of establishing communication between dissimilar networks will now be considered in greater detail. Throughout the following discussion and the description of the invention presented later in this document, the following interworking scenarios will be considered in detail:

a) interworking between the Public Switched Telephone Network (PSTN) and the Public Land Mobile Network (PLMN) and
b) interworking between the Integrated Services Digital Network (ISDN) and the Public Land Mobile Network (PLMN).

Furthermore, particular reference will be made to the provision of circuit switched telecommunications services between PSTN and PLMN networks as well as between the ISDN and PLMN networks. It should be noted that in the following, the term PSTN refers to a purely analogue land-line telephone network, while the terms ISDN and PLMN refer to networks employing entirely digital communications technology. However, although the ISDN and PLMN networks are both considered as purely digital in nature, it should be noted that in certain situations end-to-end digital communication between ISDN and PLMN networks may not be possible. This situation may arise, for example due to the characteristics of intervening connecting network elements. Thus, special consideration will also be given to ISDN - PLMN interworking issues.

When setting up a circuit switched connection for a telecommunications service, an appropriate bearer service must be established. A bearer service is a type of telecommunications service that provides the capability for the transmission of signals between user-network interfaces [Ref. ITU-T I.112]. In order for circuit switched communication to take place between two dissimilar networks, an end-to-end connection must be established. This requires an appropriate bearer service to be set up and maintained between each user's terminal equipment and its respective network interface. If the networks in question employ different communication protocols, as is generally the case, protocol conversion is also required to enable end-to-end communication.

The type of bearer service to be established depends on the type of telecommunications service required by the end users. For example, in some situations it may be necessary to establish a bearer service appropriate for a conventional speech telephone call, while in other cases a bearer service suitable for fax or data communication may be required. When establishing communication between two dissimilar networks, end-to-end indication of the type of communication requested is desirable in order to enable selection of an appropriate bearer service in both networks. As data communications and signalling protocols are typically different in the two networks, the signalling of

requested call type information between the networks cannot be achieved in a straightforward manner. This is particularly true in situations where end-to-end digital communication cannot be effected, as is the case when interworking between the analogue Public Switched Telephone Network, where call type information is provided by in-band signalling and the Public Land Mobile Network, where call type information is indicated using digital out-band signalling.

Figure 1 illustrates the architecture of the GSM Public Land Mobile Network. As can be seen from Figure 1, the GSM architecture comprises a gateway mobile switching centre (MSC) 10 which interfaces with fixed networks such as the PSTN and / or ISDN 15 and a GSM radio network. The GSM radio network comprises base station systems comprising a base station controller 16 and base transceiver stations (BTS) 17. The BSS comprises a transcoder TC (e.g. for converting to GSM speech). In Figure 1, the TC 101 is shown in the BSC, but could alternatively be in another part of the BSS, such as a BTS 17. Mobile stations are coupled to the BTSs via an air interface. The gateway MSC is also connected to subscriber and terminal equipment databases in the form of a home location register (HLR) 12, visitor location register (VLR) 13, and equipment identity register (EIR) 14. The EIR contains information relating to the mobile terminals and the VLR provides a local store of all the information required to handle calls to and from mobile users in the location area relating to that particular VLR. The HLR 12 permanently stores all the user parameters of the mobile stations, including the subscriber numbers associated with a particular mobile station and their corresponding service type. Since the PSTN is an analogue network and the GSM PLMN is digital, they are not directly compatible. Hence, the gateway MSC 10 has an associated interworking function (IWF) 11, which is a functional entity enabling interworking between the PLMN and PSTN / ISDN. Typically, the interworking function comprises a modem or modem pool for interfacing with analogue PSTN networks.

In the example considered here, the interworking function IWF in the gateway MSC 10 acts as the network interface for both the PSTN / ISDN and the PLMN and provides the necessary protocol conversion to enable end-to-end communication.

Conventionally in PLMN systems, different telecommunication services have been identified by adopting a multiple numbering scheme, that is, by dedicating a specific subscriber number (MSISDN) to a specific telecommunications service. For example, in GSM, the subscriber may currently have assigned mobile terminated GSM speech, data and / or fax numbers depending upon his profile. As previously described, there is currently considerable interest in the provision of additional and more widely varied telecommunications services. In the future, it is proposed that mobile telephone subscribers will also have the option of further high-speed data and multimedia services which, too, will have dedicated subscriber numbers.

The provisions of the current GSM multiple numbering scheme are laid down in GSM technical specification 09.07. According to GSM 09.07, a subscriber is allocated a number of MSISDNs, each associated with a particular telecommunications service. Some interworking information (IWI) is linked with each of these numbers and stored in the subscriber's HLR. The contents of the interworking information is specified in GSM 03.08 and comprises either one or two complete BC (Bearer Capability) information elements, whose contents are specified by GSM recommendation 07.01 and which are coded as described in GSM recommendation 04.08 para 10.5.4.5.

Interworking between the PSTN and a PLMN will now be described in the context of the conventional multiple number scheme as set down in GSM 09.07 and with reference to Figure 2 of the appended drawings.

As illustrated in Figure 2, when a call originates from the PSTN 15 (in this example a data call), an initial SETUP message is sent which includes the

called line identification (CLI) [Step 1]. As described above, the HLR contains a database entry corresponding to this CLI indicating the call type. The MSC 10 asks the HLR 12 for the call type and bearer capability corresponding to the CLI, and sends a setup message to the MS 18 to inform it of the incoming call and of the call type [Step 2]. The MS 18 responds by sending a CALL-CONF message after having checked its compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. This message is forwarded as a CALL-PROC message to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and a connection appropriate for the data call is made.

Use of a multiple numbering scheme, such as that defined in GSM 09.07, eliminates the need for end-to-end signalling of call type and bearer capability information in situations where end-to-end digital communication cannot be effected. This feature of the multiple numbering scheme is particularly advantageous in mobile terminated calls that originate from a PSTN. Because the mobile subscriber is allocated a separate telephone number for each telecommunications service and appropriate interworking information is associated with each number, there is no need to transmit information about the call type and required bearer services across the boundary between the PSTN and the GSM network. This approach solves the problem of how to provide information about the required bearer service for data calls originating from the PSTN. The multiple numbering scheme may also simplify interworking with ISDN networks. Although end-to-end digital communication is in theory possible between PLMN and ISDN networks, it is sometimes not possible at all levels due to properties of the ISDN or other connecting networks.

It should be noted, however, that the multiple numbering scheme defined by GSM 09.07 also has significant disadvantages. It increases the complexity of numbers to be dialled, increases the amount of subscriber numbers in use and places a burden on the user who must remember which number corresponds to which telecommunications service. Additionally, the storage of information concerning bearer capability in the HLR increases the storage requirements of the HLR, which is undesirable, particularly in view of the likely increase in the number of available telecommunications services in the near future. Furthermore, under the multiple numbering scheme, optimisation of bearer services (e.g. data transfer rates) is not possible as no mechanism for end-to-end negotiation is provided. For example, when a PSTN originated, mobile terminated call is established, the bearer service for that telecommunications service is set up with reference to the interworking information (IWI) stored in the HLR and no end-to-end adjustment of bearer services is possible. A similar situation may occur when interworking between certain ISDN networks and the PLMN, particularly when end-to-end digital connection at all levels cannot be achieved. This may lead to non-optimal use of the available transmission bandwidth in certain situations, for example if the mobile subscriber roams to a PLMN network which has different bearer capabilities from his/her home network.

Embodiments of the present invention provide a mechanism which identifies and maps call identification information (e.g. call type and bearer capability information) between dissimilar networks. Such a mapping scheme can be used for example to: a) enable optimisation of bearers in a multiple numbering scheme; b) allow the use of a reduced numbering scheme; or c) enable implementation of a single numbering scheme. In order to enable use a single numbering scheme, in which calls of all types are directed to a single subscriber number, mapping of call identification information between dissimilar networks is needed. This is achieved in embodiments of the invention by enhancing the functionality of the interworking function between networks, enabling the reformatting of call type information and enhanced

transmission of bearer capability information between networks. The mapping of the call type information eliminates the need for the aforementioned multiple numbering scheme. It thus simplifies operability of the system from a user standpoint, as the caller only needs to remember one number for the party to be called. Also, it reduces the need to increase the length of subscriber numbers to cater for additional services subscribed to in the future. Furthermore, the processing strain on the HLR is reduced. Mapping the bearer services, on the other hand, provides the additional advantage of enabling the radio bearer to be optimised. That is, the radio bearer may be negotiated to match that of the PSTN / ISDN, thus optimising the bandwidth used by the wireless network.

According to one aspect of the present invention, there is provided a switch for establishing a call between a terminal of an originating network and a terminal of a terminating network, the switch comprising: an input for receiving call identification information in a first format from the originating network; means for reformatting received call identification information into a second format; an output means for outputting the call identification information in the second format over the terminating network; and connection means for completing a connection, suitable for the identified call between the terminals.

This switch enables the elimination and / or reduction of the use of the aforementioned multinumbering scheme.

The call identification information may comprise call type information and/or bearer information.

The call type information may, for example, relate to telecommunications service information (e.g. H.324, teleservice information).

The provision of bearer information enables the terminating network to use the same bitrate as is available on the originating side, thereby ensuring that bandwidth efficiency is optimised.

The terminating network is preferably a wireless communications network, such as an UMTS or GSM network.

Optionally the switch is an MSC or GMSC.

The switch may further comprise means, coupled to the input, for determining primary call type information on the basis of a subscriber number, for forwarding first primary call identification information to the output, and for forwarding further primary call identification information to the reformatting means. Such a switch may, for example, enable a dual numbering system to be adopted, to distinguish between calls of primary types "speech" and "data". In this case, all calls of "speech" primary type are preferably automatically connected, and further determination of actual data type is made of calls of "data" primary type.

According to another aspect of the present invention, there is provided method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising: receiving call identification information in a first format from the originating network; reformatting received call identification information into a second format; outputting the call identification information in the second format over the terminating network; and completing a connection, suitable for the identified call, between the terminals.

According to a further aspect of the present invention, there is provided a method for establishing a call between a terminal of an originating network and a terminal of a terminating network, the method comprising: establishing a call of a predetermined type; transmitting call identification information in a

first format from the originating terminal to the terminating network; reformatting received call identification information into a second format; transmitting the call identification information in the second format to the terminating terminal; and establishing a connection, suitable for the identified call, between the terminals.

According to another aspect of the invention, there is provided a switching system for establishing a call from a terminal of an originating network and a terminal of a terminating network, the switching system comprising: means for receiving call identification information in a first format from the originating network; means for reformatting received call identification information into a second format; means for transmitting the call identification information in the second format over the terminating network; and connection means for completing a connection, suitable for the identified call, between the terminals.

According to an embodiment of the present invention an MSC is provided with a transcoder and interworking function. This is in contrast to conventional GSM systems for example, in which the transcoder forms part of the base station subsystem.

Embodiments of the present invention will now be described, by way of example, with reference to the accompanying drawings, of which:

Figure 1 illustrates the current GSM PLMN and its connection to the PSTN / ISDN;

Figure 2 illustrates data call establishment from a PSTN terminal to a mobile station using the conventional multiple numbering scheme;

Figure 3 illustrates establishment of an H.324 transparent video call from a PSTN terminal to a mobile station using an enhanced multiple

numbering scheme according to a first embodiment of the invention;

Figure 4 illustrates establishment of an H.324 transparent video call from a PSTN terminal to a mobile station using an enhanced multiple numbering scheme according to the first embodiment of the invention, illustrating involvement of the PSTN local exchange during call setup;

Figure 5 illustrates identification of the telecommunications service required for a PSTN originating call by an MSC of the invention;

Figure 6 illustrates establishment of a data call from a PSTN terminal to a mobile station using a single numbering scheme according to a second embodiment of the invention;

Figure 7 illustrates establishment of a data call from a PSTN terminal to a mobile station using a single numbering scheme according to a second embodiment of the invention and shows the involvement of the PSTN local exchange;

Figure 8 illustrates an alternative method of data call establishment from a PSTN terminal to a mobile station using a single numbering scheme according to the second embodiment of the invention;

Figure 9 illustrates establishment of a data call from a mobile station to a PSTN terminal using V.8bis signalling according to the second embodiment of the invention;

Figure 10 illustrates establishment of a data call from a mobile station to a PSTN terminal using V.8 signalling, according to the second embodiment of the invention;

Figure 11 illustrates establishment of a data call from a mobile station to a PSTN terminal using V.8 signalling, according to a second embodiment of the invention and shows the involvement of the PSTN local exchange;

Figure 12 illustrates establishment of a data call from an ISDN terminal to a UMTS mobile station according to the second embodiment of the invention using end-to-end digital communication;

Figure 13 illustrates establishment of a data call from an ISDN terminal to a UMTS mobile station according to the second embodiment of the invention in a situation where end-to-end digital communication cannot be effected;

Figure 14 illustrates establishment of a data call from a UMTS mobile station to an ISDN terminal according to the second embodiment of the invention using end-to-end digital communication;

Figure 15 illustrates establishment of a data call from a UMTS mobile station to an ISDN terminal according to another embodiment of the invention in a situation where end-to-end digital communication cannot be effected;

Figure 16 illustrates establishment of a data call from an ISDN terminal to a UMTS mobile station according to the second embodiment of the invention and shows the involvement of the ISDN local exchange;

Figure 17 illustrates establishment of a data call from an ISDN terminal to a UMTS mobile station according to the second embodiment of the invention in a situation where end-to-end digital communication cannot be effected;

Figure 18 illustrates establishment of a data call from a UMTS mobile station to an ISDN terminal according to the second embodiment of the invention and shows the involvement of the ISDN local exchange;

Figure 19 illustrates establishment of a data call from a UMTS mobile station to an ISDN terminal according to the second embodiment of the invention in a situation where end-to-end digital communication cannot be effected;

Figure 20 illustrates establishment of a data call between two UMTS mobile stations, according to the second embodiment of the invention;

Figure 21 illustrates establishment of a data call between a UMTS mobile station and a 2nd generation GSM mobile station according to the second embodiment of the present invention.

Figure 22 illustrates establishment of a data call from a PSTN terminal to a mobile station according to a third embodiment of the present invention;

Figure 23 illustrates, in further detail, establishment of a data call from a PSTN terminal to a mobile station according to the third embodiment of the present invention; and

Figure 24 illustrates typical mapping information for an H.324 call;

Detailed Description of the Invention

The invention concerns a method and apparatus for the control of different call types between incompatible networks, using unique call type differentiation. It is described with reference to the networks shown in Figure 1, namely the analogue PSTN network which uses in band signalling and the wireless digital GSM network which uses out band signalling. However, the invention is not restricted to call control over these networks, and is equally applicable to other non-directly compatible networks. One area in which the invention is proposed for use in the future is between a landline network (such as the PSTN or ISDN) and universal mobile telecommunication system (UMTS).

In the description of the embodiments of the invention presented later in the text, examples of call establishment between fixed line (PSTN / ISDN) and mobile (PLMN) multimedia terminals are presented. However, it should be appreciated that the invention is in no way limited to this exemplary application and may be applied, in principal, to any other kind of call establishment between dissimilar networks.

In the following, it is assumed that the fixed (PSTN / ISDN) and mobile (PLMN) multimedia terminals referred to in connection with exemplary embodiments of the invention are implemented according to ITU-T recommendations H.324, H.324/I (H.324 Annex D) or H.324/M (H.324 Annex C). The H.324 recommendations provide a framework for the implementation of multimedia terminals, permitting the communication of real-time video, audio or data, or any combination thereof. Multimedia terminals implemented according to H.324 may be stand-alone terminal equipment connected to a fixed-line PSTN / ISDN or may be mobile terminals for use in a PLMN. An H.324 multimedia terminal may also be implemented in a PC or computer workstation.

The basic H.324 recommendation sets down the requirements for a multimedia terminal designed for connection to a fixed line analogue telecommunications network, such as a PSTN telephone system. The H.324/I recommendation (H.324 Annex D) defines equivalent requirements for multimedia terminals connected to fixed line networks providing digital communication e.g. ISDN networks and H.324/M (H.324 Annex C) provides certain modifications to H.324, specifically designed to improve the robustness of the multimedia bit-stream to data transmission errors. These modifications have greatest effect on the H.223 multiplexing protocol. The suffix "M" for "mobile" is used to denote the fact that the H.324/M variant of H.324 is especially suitable for use in mobile applications, where transmission of multimedia data takes place over particularly error-prone communication channels. However, it should be noted that the use of recommendation H.324/M is not limited to mobile multimedia applications and may also be used in fixed-line networks. The differences between H.324 and H.324/M are described in Annex C of the H.324 recommendation.

While the reader is referred to the H.324, H.324/I and H.324/M recommendations for further details concerning the implementation of multimedia terminals, a brief outline of the functional elements of H.324,

H.324/I and H.324/M multimedia terminals is presented here to aid understanding of the invention.

The basic functional elements of an H.324 terminal intended for connection to an analogue PSTN are:

a) A video codec implemented according to ITU-T recommendation H.261 or H.263;
b) An audio codec according to recommendation G.723.1;
c) Data protocols to support data applications such as electronic whiteboards, still image transfer, file exchange, database access, audiographics conferencing, remote device control, network protocols etc. According to H.324 all data protocols are optional, but may include:
    i) T.120 point-to-point and multi-point teleconferencing;
    ii) T.84 point-to-point still image transfer;
    iii) T.434 point-to-point file transfer;
    iv) H.224 / H.281 far-end camera control;
    v) T.30 facsimile transfer;
    vi) T.140 text conversation protocol;
d) A control protocol according to ITU-T recommendation H.245, providing signalling to enable correct end-to-end operation of the multimedia terminal;
e) A multiplexing protocol according to H.223, for multiplexing video, audio data and control information into a single bit-stream for transmission and for demultiplexing received multimedia bit-streams; and
f) A modem, conforming to recommendations V.34 and V.8 (and optionally V.8bis).

The V.34 / V.8 / V.8bis compliant modem of an H.324 multimedia terminal is an essential element, enabling the terminal to provide an analogue output for transmission to the PSTN and enabling the conversion of analogue signals

received from the PSTN into a digital bit-stream for further processing and / or reproduction within the terminal. However, the modem may be implemented as an external unit, in which case it is connected as an interface between the multimedia terminal and the PSTN and its operation is controlled by the multimedia terminal according to recommendation V.250 (ex V.25ter).

When setting up e.g. a video call between two H.324 multimedia terminals, connected via a PSTN telephone network, the calling terminal initiates a call start-up procedure. The calling terminal first requests a connection according to the standard procedures for analogue (voice) telephony. Upon successful setup of a conventional analogue (voice) connection, the H.324 terminal enters a V.8 start-up procedure. The calling terminal transmits a CI (Call Indication) calling tone in which it signals the "H.324" V.8 CF (Call Function) which, according to the V.8 recommendation has a value 0x21. The CF information element identifies the call type required by the calling terminal. If the calling terminal subsequently detects a response from a V.34 compliant modem at the receiving end, the start-up procedure for that modem is followed. It should be noted that if both the calling and called multimedia terminals support V.8bis, the users have the opportunity to speak before proceeding to multimedia telephony.

At the end of the V.34 modem setup and handshaking procedure, a data connection is effectively established between the two multimedia terminals. Following this, system-to-system communication is initiated using the H.245 control protocol and logical channels are opened for the various information streams to be transferred.

The data rates used in end-to-end communication are defined in the V.34 recommendation and range from 2400 bits/s to 33600 bits/s. Support of the two highest data rates, 31200 bits/s and 33600 bits/s, is optional, the highest rate for which support is mandatory being 28800 bits/s. In multimedia (e.g. video) communications, higher data rates are preferred due to the large data

throughput required for real-time video. Thus, data rates of 28800 bits/s and above are considered the most appropriate for use in video telephony applications.

In an H.324/I terminal, intended for direct connection to a fixed-line digital network such as an ISDN, the V.8 / V.8bis / V.34 modem is replaced with a I.400-series ISDN user-network interface. Call setup between two H.324/I compliant multimedia terminals connected via an ISDN network is generally performed according to recommendation V.140. Call setup according to V.140 has three phases. In phase 1, the calling terminal transmits a repeating characteristic 80-bit pattern to indicate to the called terminal that it supports V.140. If a characteristic V.140 bit pattern is also detected in received data, this indicates that the called terminal supports V.140 and call setup can proceed. If no V.140 response is received after a certain predefined time-out interval, the calling terminal may fall back to any other non-V.140 protocols it supports, for example V.8 or V.8bis. In order to enable fall-back to V.8 or V.8bis, the calling terminal also transmits V.8 / V.8bis messages coded as digital audio (according to recommendation G.711) and listens for corresponding digitally encoded V.8 / V.8bis responses from the called terminal.

Assuming that a V.140 response is received during phase 1, phase 2 of the V.140 call setup procedure is entered. The calling terminal now determines the characteristics of the ISDN connection. A variety of ISDN networks are currently in use, including networks that provide communication at 56 kbits/s and 64 kbits/s. Thus, the nature of the end-to-end digital link, including data rate and bit alignment, must be confirmed before the link can be used for multimedia communication. Once the nature of the digital link has been established, phase 3 of the call setup procedure is entered. In this phase, the terminals exchange mode capabilities and select a mode. H.324/I specifies that the calling multimedia terminal provides call type information in BC (Bearer Capability) and LLC (Low Layer Compatibility) information elements.

These information elements are defined in ITU-T recommendation Q.931, which describes the protocol used in ISDN networks to establish, maintain and terminate network connections.

Once call setup using V.140 is complete, system-to-system communication between the terminals is initiated using the H.245 control protocol, just as in communication between H.324 terminals connected to a PSTN. The data rate used in end-to-end communication depends on the type of ISDN network connecting the two multimedia terminals. As stated above, ISDN networks providing data rates of 56 kbits/s and 64 kbits/s are currently in existence. Provisions are also laid down in recommendation V.110 which allow the use of lower data rates over ISDN networks. The data rates specified in V.110 correspond to many of rates used in communication between V.34 modems and are intended to provide compatibility between multimedia terminals connected to ISDN and PSTN networks.

In mobile a multimedia terminal implemented according to H.324/M (i.e. according to Annex C of H.324), the V.8 / V.34 modem is replaced with any appropriate wireless interface. H.324/M envisages that direct establishment of multimedia calls will be possible due the digital connection provided between terminals resident in PLMN networks. The data rates used for multimedia communication in future networks will be compatible with ITU-T recommendation V.110 and support will be provided for higher rates, such as 64 kbits/s.

As described above, multimedia terminals implemented according to recommendations H.324, H.324/I and H.324/M are adapted to employ different call setup procedures and provide their bearer capability information in different ways. Different data transfer rates and communications protocols are also used. Many of these differences arise from the characteristics and requirements of the networks in which the multimedia terminals are designed to operate.

The exemplary embodiments of the invention presented below describe a number of alternative procedures that enable improved call establishment between dissimilar networks, with particular emphasis on multimedia (video) telephony. These include embodiments that provide an enhanced multiple numbering scheme, embodiments that allow the use of a reduced numbering scheme and embodiments that enable implementation of a single numbering scheme. Some embodiments relate specifically to interworking between PSTN and PLMN networks, while others are related to interworking between ISDN and PLMN networks. In all cases, both mobile terminated and mobile originated call setup scenarios are described.

It assumed throughout, that in order to enable comprehensive interworking between PSTN, ISDN and PLMN networks, the interworking function of the gateway MSC has a modem or modem pool providing compatibility with V.8, V.34, V.110, V.140 and optionally V.8bis. However, no limitation as to the functionality of the IWF should be implied.

It should also be noted that the basic principles of call type and bearer service identification and mapping, as well as the call setup procedures presented in the following embodiments of the invention, apply equally well to currently existing (2^{nd} Generation) and future (3^{rd} Generation) mobile networks. Throughout the text, the current GSM (Global System for Mobile Communications) is used as an example of a 2^{nd} Generation PLMN network and the proposed UMTS (Universal Mobile Telephone System), currently undergoing standardisation, is used to exemplify a 3^{rd} Generation PLMN.

In the example interworking scenarios described below, call setup signalling within the PLMN network (i.e. between the gateway MSC and the mobile terminal) is described with reference to standardised call setup messaging defined in GSM recommendation 04.08 para 9.3. Call setup signalling within the UMTS mobile network will be closely based on the provisions of GSM

04.08 and is defined in ITU-T recommendation 24.08. Therefore close correspondence between the two systems is assumed. Furthermore, call setup signalling within the ISDN network (i.e. between ISDN terminal equipment and ISDN local exchanges) takes place in an entirely analogous manner, as defined in ITU-T recommendation Q.931.

While the reader is referred to the appropriate ITU-T recommendations for details of GSM / UMTS and ISDN call setup signalling, a summary of some important features of call setup signalling according to GSM 04.08 (version 5.3.0) are provided here in order to aid understanding of the invention. Particular emphasis is given to the information content of signals exchanged between the network and the mobile station during call setup, both in the mobile terminated (MT) and mobile originated (MO) directions. More specifically, the function and information content of the SETUP and CALL-CONF (call confirmed) will be discussed. Analogous call setup messages are used in ISDN and UMTS networks and are defined in ITU-T recommendations 24.008 and Q.931, respectively.

SETUP

In mobile terminated (MT) call setup, a SETUP message is sent by the network (e.g. from an MSC) to the mobile station in order to initiate call establishment. In mobile originated (MO) call setup, call establishment is initiated when a SETUP message is sent from the mobile terminal to the network (GSM 04.08, paragraph 9.3.23). The SETUP message contains a number of information elements (IEs) that provide call routing and control information, as well as information about the nature of call to be established. Specifically, the SETUP message may contain up to two Bearer Capability (BC) information elements (referred to as Bearer Capability 1 and Bearer Capability 2), two Low Layer Compatibility (LLC) information elements and two High Layer Compatibility (HLC) IEs.

The purpose of the BC information element(s) is to describe a bearer service(s) for a call to be established and it is used for compatibility checking as described in GSM 04.08 Annex B. Details of the structure, contents and coding of the BC IE(s) are provided in GSM 04.08 paragraph 10.5.4.5.

According to GSM 04.08, paragraph 10.5.4.18, the purpose of the Low Layer Compatibility information element is to provide a means which should be used for compatibility checking by an addressed entity (e.g. a mobile terminal in an MT call). Details of the role played by LLC IEs in compatibility checking are presented in Annex B of GSM 04.08. As will be described later in the text, according to some embodiments of the invention, the LLC IE is chosen to contain information about a requested call type (e.g. an H.324 multimedia / video call). It should be noted that when a call is established between mobile terminals within a PLMN, the LLC information element is transmitted transparently from the calling party to the called party. However, when the calling party resides in another network, for example an ISDN, there is no guarantee that end-to-end transmission of LLC information elements is possible and thus end-to-end communication of call type information in LLC IEs is not assured. Therefore, the invention provides means for transmitting call type information between networks in situations where end-to-end communication of LLC information elements is not possible .

The High Layer Compatibility information element is also intended to provide a remote user with a means for compatibility checking. The content and coding of HLC IEs is described in GSM 04.08, paragraph 10.5.4.16 and its use in compatibility checking is presented in Annex B of that recommendation. According to some example embodiments of the invention described later in the text, HLC IEs are used to contain information about a requested call type. However, as was the case for LLC IEs, while High Layer Compatibility information elements are transmitted transparently between originating and terminating parties within a PLMN, end-to-end transmission of HLC IEs between dissimilar networks cannot be guaranteed. Thus, in those

embodiments of the invention where call type information is provided in HLC information elements, means are also provided to ensure that call type information can be communicated in situations where end-to-end transmission of HLC IEs is not possible.

CALL-CONF

According to GSM 04.08, paragraph 9.3.2, a CALL-CONF (Call Confirmed) message is sent by a called mobile station as a reply to the network in order to confirm an incoming call request. Like the SETUP message, the CALL-CONF message contains a number of information elements. Specifically, it may contain up to two Bearer Capability (BC) information elements. However, BC IEs are only included in the CALL-CONF message under certain conditions. In the context of the invention, the situations in which bearer capability information is included in a CALL-CONF message are a) when the mobile station wishes a bearer service to be established other than that indicated in the Bearer Capability 1 information element of an incoming SETUP message or b) when the Bearer Capability 1 information element received in connection with an incoming SETUP message is missing (e.g. empty) or not fully specified. Examples of the use of bearer capability information in CALL-CONF messages when interworking between dissimilar networks are presented in the examplary embodiments of the invention which follow.

Establishment of a video call from a PSTN multimedia terminal to a mobile multimedia terminal (mobile terminated call) according to a first embodiment of the invention will now be described. The first embodiment of the invention relates to implementation of an enhanced multiple numbering scheme and will be described in the context of video call establishment from a PSTN multimedia terminal to a mobile multimedia terminal, as illustrated in Figures 3 and 4.

Figure 3 illustrates an example of video call establishment from a PSTN terminal to a mobile station using an enhanced multiple numbering plan.

The called party bearer capability information related to the called subscriber number is retrieved from the HLR. After reaching the connect phase [Step 8], the IWF modem starts listening to the CI (Call Indication) messages and proceeds to handshake with the modem of the originating end [Step 9]. As previously described, a standard compliant PSTN H.324 application will send the V.8 CF code 0x21 [Step 10]. Only after detecting the validity of incoming V.8 code does the MSC know that originating end is requesting a H.324 service and is not, for example a misdialled speech call to the called party's dedicated H.324 video service number.

Figure 4 illustrates the same embodiment of the invention as described in connection with Figure 3, showing the role played by the PSTN local exchange (LE) in call setup. The term local exchange is used to denote e.g. the telephone exchange with which the PSTN multimedia terminal is connected. Call setup signalling between exchanges within the PSTN network and between PSTN exchanges and (gateway) MSCs of the PLMN network takes place via a signalling network known as Signalling System Number 7 (SS7), the properties and implementation of which are defined in the ITU-T Q.700 series of recommendations. Generally, signalling between ISDN networks and the PLMN also takes place via an SS7 network, as does intra-PLMN signalling, for example between MSCs. The call setup messages used in the SS7 signalling system are termed ISUP (ISDN User Part) messages. Definitions of the ISUP messages discussed in connection with embodiments of the present invention can be found in ITU-T recommendation Q.762.

Referring to Figure 4, establishment of a video call from a PSTN multimedia terminal according to the first embodiment of the invention commences when the PSTN local exchange (LE) issues an ISUP Initial Address Message (IAM) [Step 1]. The IAM initiates seizure of an outgoing circuit and transmits number

and other information relating to routing and handling of the call. In this embodiment of the invention, in which a multiple numbering scheme is used for mobile terminated calls, the number indicated by the IAM is the called party's video service number. When the IAM is received by the MSC 10, it retrieves the call type and bearer capability information for the number indicated from the HLR 12 and transmits that information in a SETUP message to the MS 18 [Step 2]. In this case, as illustrated in Figure 4, the call type and bearer capability information retrieved from the HLR is carried in the Bearer Capability 1 information element of the SETUP message.

The MS 18 responds by sending a CALL-CONF message to the MSC 10 after having checked its compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. This message is forwarded as an ISUP Address Complete Message (ACM) to the calling terminal's PSTN local exchange [Step 4]. The ACM informs the PSTN LE that all the address signals required for routing the call to the called party have been received. Next, the MS 18 sends an ALERT message to the MSC 10, informing it that ringing has started to the called subscriber [Step 5]. This information is passed back to the PSTN LE as an ISUP Call Progress Message (CPG) [Step 6] and the PSTN LE subsequently connects the ringing tone to the calling terminal. When the called subscriber answers the call, a CONNECT message is sent from the MS 18 to the MSC 10 [Step 7]. This is transmitted back to the PSTN LE as an ISUP Answer Message (ANM) [Step 8].

After reaching the connect phase [Step 8], the IWF modem starts listening [Step 9] for the V.8 CI (Call Indication) messages expected from an H.324 compliant PSTN multimedia terminal. If a V.8 CF code 0x21 is received from the calling terminal [Step 10], the IWF modem replies, indicating its compatibility with V.34. Subsequently a V.34 start-up and handshaking procedure is initiated [Step 11] and an appropriate bearer for end-to-end communication is finally selected. As previously described, the V.34 recommendation makes provisions for the use of a range of data rates from

2400 bits/s up to 33600 bits/s, of which the higher data rates are most appropriate for video telephony. In the present exemplary embodiment, a V.34 data rate of 28800 bits/s is chosen for communication in the PSTN link of the end-to-end connection. A corresponding V.110 bearer service is selected for communication in the PLMN (GSM) link of the end-to-end connection.

The enhanced multiple numbering scheme for mobile terminated calls according to the first embodiment of the invention provides the advantage that the bearer services selected for end-to-end communication are optimised through a process of negotiation in which the interworking function of the gateway MSC listens to and interprets signalling from the originating network.

In the following, a second embodiment of the invention that employs a single numbering scheme, in which mobile subscribers are allocated a single MSISDN, will be described. In order to enable implementation of a single numbering scheme in different connection scenarios (including those where end-to-end digital connection is not possible), the gateway MSC 10 is provided with additional functionality, as outlined in Figure 5. Generally, this functionality is provided by the IWF and optionally the transcoder TC. However, alternatively, it may be provided by other parts of the MSC and / or other switches in the network.

In PSTN originated, mobile terminated call setup, the signalling detector 41 (comprising the TC and IWF) firstly detects signalling from the PSTN [Step 301]. Then it interprets the in-band signalling messages relating to the telecommunication services and maps them into appropriate digital out-band signalling [Step 302]. After mapping, the MSC 10 provides the appropriate call setup messaging for the identified call type [Step 303] to the mobile terminal. Consequently, the mobile station 18 is informed of the call type and required bearer service and an appropriate call connection is affected. Typical mapping information for an H.324 call is shown in Figure 24, and further

information on the mapping process is provided under the heading "Mapping" later in the text.

In mobile originated, PSTN terminated call setup, the MSC 10 receives out-band digital signalling from the mobile terminal indicating the type of call to be established and the bearer service(s) required. The MSC then converts (maps) the call type and bearer service information received from the MS 18 into appropriate analogue in-band signalling that can be interpreted by the terminating PSTN terminal.

Similar operations are necessary when interworking between PLMN and ISDN networks. In this case, the IWF of the MSC is generally required to map call type and bearer service information between different out-band digital formats. However, as explained above, end-to-end digital connection between PLMN and certain ISDN networks may not be possible and thus in some cases it is necessary to transmit call type information to a terminating ISDN network in-band or to receive in-band signalling from an ISDN network. According to the invention, the gateway MSC is also provided with the functionality to perform call type and bearer service information mapping operations between the PLMN and ISDN.

It should be appreciated, that depending on the exact network configuration, a given gateway MSC may enable interworking between the PLMN and PSTN only, the PLMN and ISDN only, a combination of the two, or may also provide interworking with other networks. Therefore, no limitation as to the general functionality of such a gateway MSC should be implied from the following description of the invention. Furthermore, the use of specific information elements (e.g. bearer capability BC, High / Low Layer Compatibility HLC / LLC etc.) to indicate call type and bearer capability during call setup, as well as the choice of specific values for particular fields within those information elements should not be construed as limiting the scope of the invention. The underlying characteristics of the invention apply equally

well to situations in which different information elements, data fields and values are used.

Examples of call setup according to the second embodiment of the invention will first be described by considering interworking between PSTN and PLMN networks. Two alternative procedures for mobile terminated call establishment are described with reference to Figures 6, 7 and 8. Two alternatives for establishing mobile originated calls are then described with reference to Figures 8, 9 and 10.

Figure 6 illustrates call establishment from a PSTN terminal to a mobile station using a single numbering plan according to the second embodiment of the invention. As mentioned above, in this embodiment, the MSC 10 comprises both the IWF and TC. Also, it is adapted to conform at least with V.8, and preferably V.8bis, so that it can recognise a call type from the V.8 / V.8bis call function information category. In particular, the MSC's modem pool includes a modem, which conforms to V.34, so as to support V.8 and optionally V.8bis. This modem acts as a signalling detector 41 for detecting and interpreting V.8 / V.8bis signalling.

When a call originates from the PSTN 15 (in this example a multimedia call), an initial SETUP message is sent from the PSTN terminal to the MSC 10 [Step 1]. The MSC 10, in turn, sends a SETUP message to the MS 18 associated with the subscriber number dialled, using GSM / UMTS call control signalling [Step 2]. In the example illustrated in Figure 6, the SETUP message informs the MS 18 of the incoming call and of a default call type: in this case speech. The MS 18 responds by sending a CALL-CONF message, after having checked its compatibility with the requested bearer capability [Step 3]. This message is forwarded as a CALL-PROC message to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber

[Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and a default speech connection is made [Step 9], and appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.8 / V.8bis signalling detector 41 of the MSC 10 interprets the PSTN originated V.8 / V.8bis signalling [Step 10]. The signalling related to the telecommunication services is mapped into corresponding GSM / UMTS signalling by the MSC 10. In this case the signalling detector 41 identifies the telecommunication service category as a multimedia (H.324) call and a 28.8kbps transparent bearer service category. In other cases, more than one bearer service may be selected: for example, one for images and one for data. If necessary, connections within the MSC (including activation of modems) are rearranged for example as in the GSM Phase 1 proposed "alternate speech and data" service. The GSM / UMTS speech bearer output to the mobile station speech channel is preferably blocked to prevent loudspeaker activation in the MS until the call type has been determined. However connection between the terminals is maintained.

Next, after successful detection of the signalling expected by the modem configuration, the modems at the MSC and in the PSTN start a handshaking process [Step 11] which results in connection with a commonly agreed data modulation rate.

Then, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth. The MSC instigates modification of the bearer using the GSM layer 3 call control protocol (GSM 04.08) and the MS 18 is informed of the requested call type and bearer [Step 12]. The MS returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the PSTN terminal and MS 18. Communication takes place in 3.1 kHz (external to the PLMN) bearer

service using the category involving the IWF modem and optimised radio bearer [Step 14].

Figure 7 illustrates PSTN originated, mobile terminated call setup according to the second embodiment of the invention, showing the involvement of the PSTN local exchange (LE), again using the example of an H.324 multimedia call. Further examples relating to the exchange of bearer capability information between the MSC 10 and the MS18 are also provided.

Referring to Figure 7, establishment of an H.324 call from a PSTN multimedia terminal commences when the PSTN local exchange (LE), with which the calling multimedia terminal is connected, issues an ISUP Initial Address Message (IAM) [Step 1]. When the IAM is received by the MSC 10, it sends a SETUP message to the MS 18 associated with the dialled number using GSM / UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the incoming call and, in this case, of a default call type: speech. Information concerning the requested bearer service for the default call type is indicated in the Bearer Capability (BC) information element.

The MS 18 responds by sending a CALL-CONF message to the MSC 10 after having checked its compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. Preferably, and as permitted by the provisions of GSM 04.08 para 9.3.2, the CALL-CONF message contains information relating to the call types and bearer services supported by the mobile station. The CALL-CONF message may contain one or two BC IEs and the order in which bearer capabilities are listed may indicate a preference for a particular choice of bearer service.

Having received the CALL-CONF message from the MS, the MSC issues an ISUP Address Complete Message (ACM) to the calling terminal's PSTN local exchange [Step 4]. The ACM informs the PSTN LE that all the address signals required for routing the call to the called party have been received.

Next, the MS 18 sends an ALERT message to the MSC 10, informing it that ringing has started to the called subscriber [Step 5]. This information is passed back to the PSTN LE as an ISUP Call Progress Message (CPG) [Step 6] and the PSTN LE subsequently connects the ringing tone to the calling terminal. When the called subscriber answers the call, a CONNECT message is sent from the MS 18 to the MSC 10 [Step 7]. This is transmitted back to the PSTN LE as an ISUP Answer Message (ANM) [Step 8], a default speech connection is made [Step 9] and appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.8 / V.8bis signalling detector 41 of the MSC 10 starts listening for PSTN originated V.8 / V.8bis signalling [Step 10]. Any received V.8 / V.bis signalling related to telecommunication services is mapped into corresponding GSM / UMTS signalling by the MSC 10. After successful detection of the V.8 / V.8bis signalling from the calling H.324 PSTN terminal, the modems at the MSC and in the PSTN terminal start a handshaking process [Step 11] which results in connection with a commonly agreed data modulation rate.

Next, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth. The MSC instigates modification of the bearer using the GSM layer 3 call control protocol (GSM 04.08 para 9.3.13) and the MS 18 is informed of the requested call type and bearer [Step 12]. The MS returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the PSTN terminal and MS 18 [Step 14].

Figure 8 illustrates an alternative PSTN originated, mobile terminated call setup procedure according to the second embodiment of the invention. As in Figure 7, involvement of the PSTN local exchange in call setup signalling is also shown. According to the example of Figure 8, establishment of an H.324

multimedia call begins when the PSTN local exchange, with which the calling multimedia terminal is connected, issues an ISUP Initial Address Message (IAM) [Step 1]. When the IAM is received by the MSC 10, it sends a SETUP message to the MS 18 associated with the dialled number using GSM / UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the incoming call, but unlike the example call setup procedure described in connection with Figures 6 and 7, no default call type is indicated to the mobile terminal. Instead, the bearer capability information element of the SETUP message is left empty. According to this alternative call setup procedure, receipt of a SETUP message containing no bearer capability information, informs the mobile terminal that the calling party resides outside the PLMN network and is not able to provide direct out-band digital signalling relating to the call type and bearer service required. The mobile station forms a CALL-CONF message and transmits it to the MSC [Step 3]. Preferably, both call type and bearer capability information is provided in the CALL-CONF message in a standardised manner in accordance with the provisions of GSM 04.08 para 9.3.2 (or equivalent UMTS recommendation). However, it should be appreciated that use of particular information elements and the exact manner in which the information is coded is not significant for application of the invention.

Having received the CALL-CONF message, the MSC issues an ISUP Address Complete Message (ACM) to the calling terminal's PSTN local exchange [Step 4]. Next, the MS 18 sends an ALERT message to the MSC 10, informing it that ringing has started to the called subscriber [Step 5]. This information is passed back to the PSTN LE as an ISUP Call Progress Message (CPG) [Step 6] and the PSTN LE subsequently connects the ringing tone to the calling terminal. When the called subscriber answers the call, a CONNECT message is sent from the MS 18 to the MSC 10 [Step 7]. This is transmitted back to the PSTN LE as an ISUP Answer Message (ANM) [Step 8].

At this point, the V.8 / V.8bis signalling detector 41 of the MSC 10 starts listening for PSTN originated in-band signalling that might indicate the call type and bearer service required by the calling terminal [Step 9]. Specifically, if the signalling detector receives a V.8 CF code 0x21, the signature of an H.324 standard compliant PSTN multimedia terminal, the IWF modem replies, indicating its compatibility with V.34. Subsequently a V.34 start-up and handshaking procedure is initiated [Step 10], which results in connection between the MSC IWF and the calling PSTN terminal with a commonly agreed data modulation rate. The bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth and the connection is completed for end-to-end multimedia communication [Step 11]. In the example described in Figure 8, a V.34 data rate of 28800 bits/s is chosen for communication in the PSTN link of the end-to-end connection and a corresponding V.110 bearer is selected for communication in the PLMN (GSM) link.

If no appropriate in-band signalling can be identified by the signalling detector 41, or the bearer capabilities of the PSTN and mobile terminal are not compatible, a speech connection is established, or alternatively, the call is terminated.

Figures 9, 10 and 11 illustrate two alternative examples of call establishment for a mobile originated, PSTN terminated multimedia call, according to the second embodiment of the invention. In the first alternative, it is assumed that a default speech call is first established and then modified to a multimedia call at some later time using V.8bis signalling generated in, and then transmitted in-band from the MSC to the PSTN multimedia terminal. In the second alternative mobile originated call setup scenario, a multimedia call is established directly using appropriate V.8 signalling, generated by the MSC and transmitted in-band to the receiving PSTN terminal. It should be appreciated that the first alternative requires both the MSC and the receiving PSTN multimedia terminal to have V.8bis functionality. If either one is not

compliant with V.8bis, mobile originated multimedia call setup proceeds according to the second alternative.

Figure 9 illustrates a first alternative method of call establishment from a GSM / UMTS mobile station to a PSTN terminal (mobile originated call) according to the second embodiment of the invention. In this alternative method, a default speech call is first established between the MS and the PSTN terminal. Then, at some later time, either at the instigation of the user, or automatically, appropriate V.8bis is sent to the PSTN terminal in order to effect the changeover from speech to multimedia call. This method of mobile originated multimedia call establishment therefore requires that the MSC 10 is adapted to conform with V.8bis, so that it can send and recognise a call type from the V.8bis call function information category. In particular, for PSTN interworking the MSC's modem pool includes a modem, which conforms to V.34, so as to support V.8bis. The modem-transcoder unit of MSC acts as a signalling source and detector 41 for detecting and interpreting V.8bis signalling.

Referring to Figure 9, when a call originates from the MS 18 (in this example a multimedia call), an initial SETUP message is sent from the MS 18 to the MSC10. In this example, it is assumed that the SETUP message includes a video service specific codepoint by default in both the bearer capability BC and HLC / LLC information elements [Step 1]. The MSC 10 interprets the requested service from the BC information element and sends a SETUP message to the PSTN terminal 15 [Step 2]. As will be described in further detail later in the text, in the situation where the called party resides within a digital call control domain (ISDN, other PLMN), the called terminal interprets the requested service from the peer-to-peer transmitted HLC / LLC information element. However, in the current case, where the receiving terminal resides in a PSTN, the SETUP message [Step 2] does not forward digital call control specific information to PSTN terminal 15 to indicate the requested bearer service. Instead, it merely indicates a default call type (e.g.

speech) using the BC element. The PSTN terminal responds by sending a CALL-CONF message, without indication of the expected bearer service [Step 3]. Receipt of a CALL-CONF message without indication of bearer capability information effectively informs the MSC that the called party resides in a PSTN network. A CALL-PROC message is forwarded to the mobile station 18 by the MSC 10 [Step 4]. Then the PSTN terminal sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the PSTN terminal to the MSC [Step 7] and then from the MSC to the MS 18 [Step 8] when the called subscriber answers and a default speech connection is made [Step 9].

At the time the mobile station wishes to start the transition to video call (immediately, or later at the wish of mobile station user), the DCE in signalling detector 41 is told by the PLMN to send appropriate V.8bis signalling (for example, as described in Intel's Videophone Ready Modem Handbook Revision 1.1 and the H.324 specification) towards the PSTN terminal 15 [Step 10]. When the appropriate signalling is interpreted at the PSTN terminal 15, it switches to the appropriate DTE-DCE mode and starts modem handshaking with the modem in MSC 10 [Step 11]. This results in establishment of a data connection between modems with a common data modulation.

Next, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type [Step 12]. The MS also gets confirmation that it has received an acknowledgement for its original video call mode change request, and returns a message to the MSC to complete the UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the MS 18 and PSTN terminal [Step 14]. Communication takes place in 3,1 kHz (external to the

PLMN) bearer service category [Ref: GSM 2.02] involving a modem in the interworking function of MSC 10 and using the optimised radio bearer.

Figure 10 illustrates a second alternative method of call establishment from a GSM / UMTS mobile station 18 to a PSTN terminal (mobile originated call setup) according to the second embodiment of the invention. More particularly, it shows establishment of an MS originated video call with direct video initialisation using MSC–configured V.8 signalling which originates in a data mode enabled modem of the MSC 10. Here it is assumed that the interworking function of the MSC is adapted to conform with V.8, (but not V.8bis), so that it can send and recognise a call type from the V.8 call function information category. In particular, the MSC's modem pool includes a modem, which conforms to V.34, so as to support V.8. This modem acts as a signalling source and detector 41 for detecting and interpreting V.8 signalling.

Referring to Figure 10, when a call originates from the MS 18 (in this example a multimedia call), an initial SETUP message is sent from the MS 18 to the MSC 10 [Step 1]. In this case, as in the example illustrated in Figure 9, it is assumed that the SETUP message sent from the mobile station to the MSC includes a video service specific codepoint by default in both the bearer capability BC and HLC information elements.

Having received the SETUP message from the mobile station, the MSC 10, in turn, sends a SETUP message to the PSTN terminal 15 associated with the dialled subscriber number [Step 2]. The SETUP message informs the PSTN terminal 15 of the incoming call, but in this case no default call type is indicated. The PSTN terminal 15 responds by sending a CALL-CONF message [Step 3], which is forwarded as a CALL-PROC message to the mobile station 18 by the MSC 10 [Step 4]. Then, the PSTN terminal sends an ALERT message to the MSC, informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Next, a CONNECT message is sent from the

PSTN terminal 15 to the MSC [Step 7] and the MSC, in turn transmits a CONNECT message to the mobile station [Step 8].

At this point, the MSC instructs the signalling detector 41 to generate in-band signalling to identify the call type and bearer service required by the calling mobile station to the PSTN terminal. Thus, in the current example, the IWF modem starts generating V.8 CI (Call Indication) messages, including the 0x21 CF code expected from an H.324 compliant multimedia terminal [Step 9]. If the signalling detector 41 subsequently receives an appropriate reply from the PSTN, for example indicating that the called terminal incorporates a V.34 compatible modem, the IWF initiates a V.34 start-up and handshaking procedure, resulting in connection between the MSC and the PSTN with a mutually agreed data modulation rate. The bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth and the connection is completed for end-to-end communication [Step 10a]. In this case using a V.34 data modulation rate of 28.8 kbits/s is established in the PSTN link and a corresponding V.110 data rate is used in the PLMN (GSM) link.

In the event that the signalling detector does not receive a reply from the PSTN, indicating that it is in communication with a V.34 compliant modem then, advantageously, after a given time-out or retry period, the MSC instigates establishment of a speech call between the MS and PSTN terminal. An end-to-end speech connection is then established [Step 10b]. This provides the system with a "fall-back" capability, preventing call failure in situations where, for example, the called PSTN does not support the call type and / or bearer service requested by the calling PLMN terminal.

Figure 11 illustrates the same mobile originated, PSTN terminated call set up scenario as described in connection with Figure 10, showing the role played by the PSTN local exchange.

Call establishment between ISDN and PLMN terminals according to the second embodiment of the invention will now be described with reference to Figures 12 to 19, which cover a variety of call setup scenarios, including both mobile terminated and mobile originated calls.

Figures 12 to 15 describe call setup between ISDN and PLMN networks assuming that a video service specific codepoint is included by default in both the bearer capability BC and HLC information elements. Figures 12 and 13 illustrate call establishment from an ISDN multimedia terminal to a mobile multimedia terminal (mobile terminated call), while Figures 14 and 15 illustrate call establishment from a mobile multimedia terminal to an ISDN multimedia terminal (mobile originated call). Figures 13 and 15 illustrate how a multimedia call can be established between the ISDN and PLMN networks when end-to-end (peer-to-peer) out-band digital signalling cannot be effected.

Figures 16 to 19 describe call setup between ISDN and PLMN networks assuming that a video service specific codepoint is included by default in both the bearer capability BC and LLC information elements. Figures 16 and 17 illustrate call establishment from an ISDN multimedia terminal to a mobile multimedia terminal (mobile terminated call), while Figures 18 and 19 illustrate call establishment from a mobile multimedia terminal to an ISDN multimedia terminal (mobile originated call). Figures 17 and 19 illustrate how a multimedia call can be established between the ISDN and PLMN networks when end-to-end (peer-to-peer) out-band digital communication is not possible.

Each of the ISDN / PLMN call setup scenarios introduced above will now be considered in detail with reference to the figures.

Figure 12 illustrates mobile terminated digital domain call setup with peer-to-peer HLC assuming global acceptance of the HLC information element to indicate an H.324 call type and use of a single numbering scheme for mobile

terminated calls. When a multimedia call originates from an ISDN terminal, an initial SETUP message is sent to the gateway MSC 10 [Step 1]. The SETUP message includes the called line identification (CLI) and an indication of the requested call type / bearer service. In this example, it is assumed that the SETUP message includes a video service specific codepoint by default in both the bearer capability BC and HLC information elements and that the ISDN terminal requests a UDI (Unrestricted Digital Information) 64 kbits/s bearer service. Assuming that end-to-end transmission of the HLC information element is possible, the MSC 10 interprets and forwards the requested bearer service and HLC information elements in a SETUP message to the mobile station 18 [Step 2]. The mobile station responds by sending a CALL-CONF message to the MSC, after having checked its compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. If the mobile station cannot support the requested bearer service, it provides an indication of an alternative bearer service, for example a V.110 28.8 kbits/s bearer, in the CALL-CONF message. This message is forwarded as a CALL-PROC message to the ISDN terminal by the MSC 10 [Step 4]. Next, the MS sends an ALERT message to the MSC, informing it that ringing has started to the called subscriber [Step 5] and subsequently the MSC connects the ringing tone to the calling subscriber [Step 6]. In the case that the requested bearer service can be supported by the MS 18, a CONNECT message is sent from the MS to the MSC when the called subscriber answers [Step 7a]. The MSC then issues a CONNECT message to the ISDN terminal [Step 8a] and a connection according to the originally requested bearer is made [Step 9a]. If an alternative bearer was suggested by the mobile terminal in the CALL-CONF message, call connection proceeds as shown in Steps 7b, 8b and 9b, resulting in the establishment of an alternative bearer.

Figure 13 illustrates call establishment from an ISDN terminal 15 to a mobile station 18 using a single numbering scheme according to the second embodiment of the invention. As in the example of Figure 12, it is assumed that the initial SETUP message transmitted from the ISDN terminal includes a

video service specific codepoint by default in both the bearer capability BC and HLC information elements. However, in this case, it is further assumed that end-to-end digital communication is not possible. As explained earlier, networks exist which fail to send out-band signalling messages end-to-end, and thus in this embodiment, the transcoder in MSC 10 is adapted to conform with V.140 (the ISDN in-band signalling protocol comparable to V.8 / V.8bis used in PSTN networks), so that it can recognise a call type from the in-band transmitted V.140 information. In particular, the MSC's modem-transcoder 41, is adapted to detect V.140 signalling in the LSBs of 56 / 64 kbps PCM modulated data as explained in the V.140 specification. Thus, the modem-transcoder unit of MSC acts as a signalling detector 41 for detecting and interpreting V.140 signalling.

When a call originates from an ISDN terminal 15 (in this example a multimedia call), an initial SETUP message is sent from the ISDN terminal to the MSC 10 [Step 1]. The MSC 10, in turn, sends a SETUP message to the MS 18 associated with the subscriber number dialled using GSM / UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the incoming call and of a default call type: in this case speech with 3,1 kHz bearer service category (GSM 02.02). The MS 18 responds by sending a CALL-CONF message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded as a CALL-PROC message to the ISDN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the ISDN [Step 8]. When the called subscriber answers and a default speech connection is made [Step 9], appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.140 signalling detector 41 of the MSC 10 listens for and interprets ISDN originated V.140 signalling [Step 10]. The detected signalling related to the telecommunication services is mapped into corresponding GSM / UMTS signalling by the MSC 10. In this case, the detector 41 identifies the telecommunication service categories as a multimedia (H.324) call and a 64kbps transparent UDI bearer service category and the MSC instigates modification of the bearer using the layer 3 call control protocol.

Then, the bearer capability for the wireless network is set to be the same as that determined for the ISDN, and the MS 18 is informed of the requested call type in the BC information element [Step 11]. The MS returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 12]. The connection is now complete for UDI multimedia communication between the PSTN terminal and MS 18 [Step 13].

Figure 14 provides two examples of mobile originated, ISDN terminated call setup, assuming peer-to-peer digital communication. In other words, it is assumed that end-to-end digital communication is possible i.e. through the ISDN, PLMN and any intervening networks, for example an SS7 network linking the ISDN local exchange and the gateway MSC. In the first example, the mobile terminal requests a UDI bearer service of 64 kbits/s and in the second a V.110 / 28.8 kbits/s bearer service is requested. In both cases, it is assumed that the called terminal is compatibile with the requested call type / bearer service.

Following the call setup scenario shown in Figure 14, when a multimedia call originates from a mobile terminal, an initial SETUP message is sent to the gateway MSC 10 [Step 1a / 1b]. The SETUP message includes the called line identification (CLI) and an indication of the requested call type / bearer service. Again, it is assumed that the SETUP message includes a video service specific codepoint by default in both the bearer capability BC and HLC

information elements. Assuming that end-to-end digital transmission of the information elements is possible, the MSC 10 interprets and forwards the requested call type and bearer service information in a SETUP message to the ISDN terminal [Step 2a / 2b]. The ISDN terminal responds by sending a CALL-CONF message to the MSC [Step 3a / 3b]. The MSC then issues a CALL-PROC message to the calling mobile station [Step 4a / 4b] to indicate that call setup is proceeding.

Next, the ISDN terminal sends an ALERT message to the MSC, informing it that ringing has started to the called subscriber [Step 5a / 5b] and subsequently the MSC connects the ringing tone to the calling subscriber [Step 6a / 6b]. Finally, a CONNECT message is sent from the ISDN terminal to the MSC when the called subscriber answers [Step 7a / 7b]. The MSC then issues a CONNECT message to the mobile station [Step 8a / 8a] and the requested bearer service is established for end-to-end communication [Step 9a / 9b].

Figure 15 illustrates call establishment from a mobile station to an ISDN terminal (mobile originated call) according to the second embodiment of the invention, in the situation that end-to-end communication of the HLC is not possible. In this embodiment, the transcoder in the MSC 10 is adapted to conform with V.140 so that it can indicate a call type as in-band transmitted V.140 information. In particular, the MSC's modem-transcoder 41 can indicate V.140 signalling in the LSBs of 56 / 64 kbps PCM modulated data, as explained in the V.140 specification. Thus, the modem-transcoder unit of MSC acts as a signalling source 41 for transmitting, detecting and interpreting V.140 signalling.

When a call originates from the MS 18, an initial SETUP message is sent which, in this example, includes the video service specific codepoint by default in both BC and HLC information elements [Step 1]. The MSC 10 interprets the requested service from the BC information element and in

optimal case the called party would reside within digital call control domain (ISDN, other PLMN) end terminal would interpret it from the peer-to-peer transmitted HLC information element. However, in a network environment which does not conform to ISUP, the SETUP message transmitted from the MSC [Step 2] does not forward the digital call control specific information to the ISDN terminal. The ISDN terminal responds by sending a CALL-CONF message [Step 3].

Thus, connection signalling is completed with knowledge by MSC 10 that the called party resides in either an ISDN or PSTN network [Steps 4 – 8]. As both H.324 speech call first and direct video service signalling could be anticipated from the MS 18, the default user data (in this case speech) is switched to pass through a transcoder-voice modem unit 41 of MSC10 [Step 9]. At the time the mobile station wishes to start transition to a video call (immediately or later at the wish of mobile station user), the transcoder in the signalling detector 41 is instructed by the PLMN to send appropriate V.140 (and / or simultaneous V.8bis signalling) as described in V.140 and Intel's Videophone Ready Modem Handbook & H.324 specification towards the ISDN terminal 15 [Step 10]. When the signalling is interpreted in the ISDN terminal, it switches to an appropriate UDI mode and activates the video application at its own end.

Next, the bearer capability for the wireless network is set to be the same as that determined for the ISDN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type within the BC information element [Step 11]. The MS also gets a confirmation that it has received an acknowledgement for its original video call mode change request. The MS 18 returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 12]. The connection is now complete for multimedia communication between the MS 18 and the ISDN terminal [Step 13]. Communication takes place in 64 kbps UDI bearer service category.

Figure 16 illustrates call establishment from an ISDN terminal to a mobile station (mobile terminated call) using a single numbering scheme according to the second embodiment of the invention and further shows the involvement of the ISDN local exchange (LE) in call setup. The example of Figure 16 once more illustrates establishment of a multimedia (video) call. In this case, it is assumed that call type information is provided by default in both the bearer capability BC and LLC information elements transmitted from the ISDN terminal to the gateway MSC in the initial call SETUP message. This is in contrast to the examples presented in Figures 12 to 15, where call type information was carried in the BC and HLC information elements. In Figure 16, it is further assumed that the ISDN and PLMN are such that the LLC information element may be communicated end-to-end between networks.

Call control signalling between the ISDN terminal and the ISDN local exchange with which it is connected is implemented according to ITU-T recommendation Q.931 which provides standardised signalling for call setup, maintenance and teardown in ISDN networks. The ISDN call control procedures defined in Q.931 are substantially similar to those used in GSM call control signalling. Furthermore, as in the case of interconnection between the PSTN and PLMN, ISDN local exchanges are general connected to gateway MSCs of the PLMN network via an SS7 network, as defined in the ITU-T Q.700 series recommendations. Call control signalling between the ISDN local exchange and the gateway MSC is accomplished using ISUP (ISDN User Part) messaging as defined in ITU-T recommendation Q.762.

Call establishment from an ISDN terminal to a PLMN mobile terminal commences when the ISDN terminal issues a Q.931 SETUP message [Step 1]. In this example, it is assumed that a video service specific codepoint is included by default in both the bearer capability BC and LLC information elements of the SETUP message. When the Q.931 SETUP message is received by the ISDN local exchange, it issues an ISUP IAM (Initial Address Message) to the gateway MSC of the PLMN [Step 2]. As end-to-end digital

signalling via the ISDN, SS7 and PLMN network is assumed possible in this example, the MSC is able to receive direct out-band signalling indicating the contents of the LLC and BC information elements. The MSC interprets the information concerning the requested call type and bearer capability received in connection with the IAM message and converts it into a format appropriate for transmission in the PLMN, for example as a GSM SETUP message, according to GSM 04.08 para 9.3.23 [Step 3] or its UMTS equivalent. Meanwhile, the ISDN local exchange informs the calling terminal that call setup is progressing by issuing a Q.931 CALL-PROC message [Step 4].

On receipt of the SETUP message, the mobile station checks its compatibility with the requested call type and bearer capability and transmits a CALL-CONF message back to the MSC [Step 5]. The MSC subsequently issues an ISUP Address Complete Message (ACM) to the ISDN local exchange to indicate that all the address signals required for routing of the call to the called party have been received [Step 6]. Next, the mobile terminal sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 7]. This information is forwarded as an ISUP Call Progress Message (CPG) via the SS7 network to the ISDN local exchange [Step 8] which, in turn, sends a Q.931 ALERT message to the calling ISDN terminal and subsequently connects the ringing tone to the calling subscriber.

When the called subscriber answers, a CONNECT message is sent from the MS to the MSC and is forwarded as an ISUP Answer Message (ANM) to the ISDN local exchange. Finally, the ISDN local exchange sends a Q.931 CONNECT message to the calling subscriber and a connection suitable for the call in question is established. If the mobile station can support the call type and bearer originally requested by the calling ISDN terminal, the call connection phase proceeds as indicated by steps 10a, 11a, 12a and 13a, of Figure 16. In the example of Figure 16, the calling ISDN terminal initially requests a multimedia (video) call using a 64 kbits/s UDI bearer service. In the event that the mobile terminal cannot provide the required bearer service

(e.g. it does not support the required data rate), this becomes apparent to the MSC at step 5, when the MS replies to the initial SETUP message. As explained earlier, according to GSM 04.08 para 9.3.2, the mobile terminal can provide alternative bearer capability information in the CALL-CONF message. Final call connection then proceeds according to steps 10b, 11b, 12b and 13b, in this example resulting in the establishment of an V.110 28.8 kbits/s bearer for the end-to-end connection.

Figure 17 also illustrates call establishment from an ISDN terminal to a mobile station (mobile terminated call) using a single numbering scheme according to the second embodiment of the invention. As in the example of Figure 16, it is assumed that a video service specific codepoint is included by default in both the bearer capability BC and LLC information elements of the SETUP message. However, in this case, it is assumed that end-to-end digital out-band communication is not possible, for example due to properties of the intervening network. In order to enable call setup using the aforementioned single numbering scheme in this situation, the interworking function IWF of the gateway MSC is invoked during call setup. Specifically, the IWF is adapted to conform with V.140, so that it can recognise a call type from in-band transmitted V.140 information. In particular, the MSC's modem-transcoder 41, is adapted to detect V.140 signalling in the LSBs of 56 / 64 kbps PCM modulated data as explained in the V.140 specification. Thus, in this case, the modem-transcoder unit of MSC acts as a signalling detector 41 for detecting and interpreting V.140 signalling.

When a call originates from an ISDN terminal 15 (again, in this example a multimedia call), a Q.931 SETUP message is sent from the ISDN terminal to the MSC 10 [Step 1]. Information concerning the requested call type and bearer service is present in the BC and LLC information elements of the Q.931 SETUP message. The MSC 10 then sends a SETUP message to the MS 18 associated with the subscriber number dialled using GSM / UMTS call control signalling [Step 2]. The SETUP message informs the MS 18 of the

incoming call and of a default call type: in this case speech with 3,1 kHz bearer service category (GSM 02.02). The MS 18 responds by sending a CALL-CONF message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded as a CALL-PROC message to the ISDN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the ISDN [Step 8]. When the called subscriber answers and a default speech connection is made [Step 9], appropriate speech transcoding is activated in the transcoder unit.

Upon connection, the V.140 signalling detector 41 of the MSC 10 starts to listen for and interpret ISDN originated V.140 signalling [Step 10]. The detected signalling related to the telecommunication services is mapped into corresponding GSM / UMTS signalling by the MSC 10 and, in this embodiment, the MSC instigates modification of the bearer using the layer 3 call control protocol. In other words, the detector 41 identifies the telecommunication service categories as a multimedia (H.324) call and a 64kbps transparent UDI bearer service category.

Then, the bearer capability for the wireless network is set to be the same as that determined for the ISDN, and the MS 18 is informed of the requested call type in the BC information element [Step 11]. The MS returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 12]. The connection is now complete for UDI multimedia communication between the ISDN terminal and MS 18 [Step 13].

Figure 18 illustrates two examples of call establishment from a mobile station to an ISDN terminal (mobile originated call) according to the second embodiment of the invention. It also illustrates the involvement of the ISDN

local exchange in call setup, showing the signalling that occurs between the gateway MSC of the PLMN network and the ISDN LE. In this case, it is further assumed that end-to-end digital communication is possible i.e. through the PLMN, ISDN and any intervening networks, for example an SS7 network linking the gateway MSC of the PLMN and the ISDN local exchange.

As in all previous cases, specific reference is made to video call establishment between multimedia terminals resident in the two networks. In the first example, illustrated by Steps 1a to 13a in Figure 18, it is assumed that the calling mobile terminal (and the PLMN) is capable of supporting video telephony using a UDI bearer at a data rate of 64 kbits/s, corresponding to the maximum data rate available for communication in the ISDN network. This situation can be considered optimal, as the capabilities of the mobile station (and the PLMN) match those of the ISDN and the maximum data rate available in the system can be used for end-to-end communication. It is assumed that mobile stations designed for use in 3rd Generation PLMN networks such as the UMTS system will be capable of supporting this kind of bearer service.

In the second example of mobile originated call setup, represented by Steps 1b to 13b in Figure 18, it is assumed that the calling mobile station and / or the PLMN does not support a UDI bearer at a data rate of 64 kbits/s. Therefore, the MS requests a video call to be established using a bearer service with a lower data rate, in this case a V.110, 28.8 kbits/s bearer service. The data rate used for end-to-end communication is thus limited by the capabilities of the mobile station and / or PLMN. This situation might arise, for example when using a first generation multimedia mobile terminal in 2nd Generation PLMN networks offering data rates appropriate for video telephony.

As shown in Steps 1a to 13a of Figure 18, call establishment from a PLMN mobile station to an ISDN terminal commences when the PLMN terminal

issues a SETUP message [Step 1a], conforming with the recommendations of GSM 04.08 para 9.3.23 or its UMTS equivalent. In the example call setup scenarios depicted in Figure 18, it is assumed that a video service specific codepoint is included by default in both the bearer capability BC and LLC information elements of the SETUP message.

The SETUP message is received by the MSC of the PLMN, which subsequently sends an ISUP Initial Address Message (IAM) to the ISDN local exchange [Step 2a]. As end-to-end digital signalling via the PLMN, SS7 and ISDN network is assumed possible in this example, the ISDN LE receives direct out-band signalling indicating the contents of the LLC and BC information elements. It then transmits a Q.931 SETUP message to the called ISDN terminal, indicating the requested call type and bearer capability in the BC and LLC information elements of the SETUP message [Step 3a]. Meanwhile, the gateway MSC informs the calling mobile station that call setup is proceeding by issuing a CALL-PROC message [Step 4a].

On receipt of the SETUP message, the ISDN terminal checks its compatibility with the requested call type and bearer capability and transmits a Q.931 CALL-CONF message back to the ISDN local exchange [Step 5a]. The ISDN local exchange subsequently issues an ISUP Address Complete Message (ACM) to the MSC to indicate that all the address signals required for routing of the call to the called party have been received [Step 6a]. Next, the ISDN terminal sends an ALERT message to the ISDN local exchange, informing it that ringing has started to the called subscriber [Step 7a]. This information is forwarded as an ISUP Call Progress Message (CPG) via the SS7 network to the MSC [Step 8a] which, in turn, sends an ALERT message to the calling mobile station [Step 9a] and subsequently connects the ringing tone to the calling subscriber.

When the called subscriber answers, a Q.931 CONNECT message is sent from the ISDN terminal to the MSC [Step 10a] and is forwarded as an ISUP

Answer Message (ANM) to the MSC [Step 11a]. Finally, the MSC sends a CONNECT message to the calling subscriber [Step 12a] and a connection suitable for the call in question is established [13a], in this case a UDI bearer service with a data rate of 64 kbits/s.

In the alternative call setup scenario illustrated by Steps 1b to 13b in Figure 18, call establishment proceeds in a manner similar to that just described. In this case, however, the mobile station does not support the 64 kbits/s UDI bearer, but indicates its support for a V.110 28.8 kbits/s bearer service in the initial SETUP message.

Figure 19 illustrates call establishment from a mobile station to an ISDN terminal (mobile originated call setup) according to the second embodiment of the invention, in the situation that end-to-end communication of the LLC information element is not possible. In this embodiment, the transcoder in the MSC 10 is adapted to conform with V.140 so that it can indicate a call type as in-band transmitted V.140 information. In particular, the MSC's modem-transcoder 41 can indicate V.140 signalling in the LSBs of 56 / 64 kbps PCM modulated data, as explained in the V.140 specification. Thus, the modem-transcoder unit of MSC acts as a signalling source 41 for transmitting, detecting and interpreting V.140 signalling. In this case, call setup proceeds in a manner analogous to that described in connection with Figure 15.

Figures 20 and 21 illustrate mobile-to-mobile call setup scenarios according to the second embodiment of the invention, again using the establishment of a multimedia call as an example. Figure 20 describes call establishment between two UMTS mobile stations, and shows the ISUP call setup signalling that takes between MSCs within the PLMN network. Figure 21 illustrates call establishment between a UMTS mobile station and a 2nd generation (e.g. GSM) mobile station.

A third embodiment of the invention enabling the use of a reduced numbering plan will now be described. Figures 22 and 23 illustrate call establishment from a PSTN terminal to a mobile station using a dual numbering plan. This embodiment provides a hybrid arrangement in which only two numbers are used: one to identify speech calls and the other to identify data calls. As in the arrangement of Figure 2, the HLR 12 contains a database entry corresponding to the called line identification indicating the call type. However, in this embodiment, this database has only two entries: one indicating that the call type is speech and the other indicating that the call type is data. If the call type is speech, the PSTN originating signal is transcoded by the TC 101 of the MSC 10 and transmitted over the radio network. However, if it is data, the actual type is determined by the IWF 11 as described with reference to Figure 5 above.

Call establishment in this embodiment will now be described in more detail, with reference to Figures 22 and 23.

Figure 22 illustrates call establishment for a speech call. When a call originates from the PSTN 15, an initial SETUP message is sent which includes the called line identification (CLI) – in this case for a speech call. [Step 1]. As mentioned above, the HLR contains a database entry corresponding to this CLI indicating the call type as speech. The MSC 10 asks the HLR 12 for this call type (bearer capability) corresponding to the CLI, and sends a SETUP message to the MS 18 to inform it of the incoming call and that the call type is speech [Step 2]. The MS 18 responds by sending a CALL-CONF message after having checked the compatibility (e.g. user profile capability) with the requested bearer capability [Step 3]. This message is forwarded as a CALL-PROC message to the PSTN by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from

the MSC to the PSTN [Step 8] when the called subscriber answers and a connection appropriate for the speech call is made.

Figure 23 illustrates call establishment for a data call. When a call originates from the PSTN 15 (in this example a multimedia call), an initial SETUP message is sent which includes the called line identification (CLI) – in this case for a data call. [Step 1]. As mentioned above, the HLR contains a database entry corresponding to this CLI indicating the call type as data. The MSC 10 asks the HLR 12 for this call type (bearer capability) corresponding to the CLI, and sends a setup message to the MS 18 to inform it of the incoming call and of a default call type (e.g. a default data type) [Step 2]. The MS 18 responds by sending a CALL-CONF message after having checked the compatibility with the requested bearer capability [Step 3]. This message is forwarded to the PSTN as a CALL-PROC message by the MSC 10 [Step 4]. Then, the MS sends an ALERT message to the MSC informing it that ringing has started to the called subscriber [Step 5] and consequently the MSC connects the ringing tone to the calling subscriber [Step 6]. Finally, a CONNECT message is sent from the MS to the MSC [Step 7] and then from the MSC to the PSTN [Step 8] when the called subscriber answers and the default connection is made [Step 9].

Next, the MSC endeavours to determine the actual data type. In this example, upon connection, the V.8 / V.8bis signalling detector 41 of the MSC 10 interprets the PSTN originated V.8 / V.8bis signalling [Step 10]. The signalling related to the telecommunication services is mapped into corresponding GSM / UMTS signalling by the MSC 10 and, in this embodiment, the MSC instigates modification of the bearer using the GSM layer 3 call control protocol (GSM 04.08 para 9.3.13). In other words, in this case the detector 41 identifies the telecommunication service as a multimedia call (H.324) and the bearer service as a 28.8kbps transparent bearer service category. If necessary, connections within the MSC (including activation of modems) are rearranged for example as in the GSM Phase 1 proposed “alternate speech

and data" service. The GSM / UMTS speech bearer output to the mobile station speech channel is preferably blocked to prevent loudspeaker activation in the MS until the call type has been determined. However connection between the terminals is maintained.

Next, modem handshaking takes place [Step 11]. The, the bearer capability for the wireless network is set to be the same as that determined for the PSTN, so as to optimise the bandwidth, and the MS 18 is informed of the requested call type [Step 12]. The MS returns a message to the MSC to complete the GSM / UMTS bearer modification process [Step 13]. The connection is now complete for multimedia communication between the PSTN terminal and MS 18 [Step 14]. Communication takes place via the MSC modem pool using the optimised radio bearer.

Mapping

Typical mapping operations performed by the MSC will now be described, firstly from V.8 to GSM and then optional mapping for V.8bis. As will be appreciated, this mapping is only exemplary and similar mapping could take place for future networks such as UMTS. GSM signalling is generally derived from ISDN, i.e. it is based on ITU-T Q.931 recommendation. So too are 3G telecommunication systems such as UMTS, hence, their particular compatibility with such mapping methods.

In this embodiment, information relating to the telecommunications service is mapped into the HLC (high level capability) information element in GSM (based on Q.931). Similarly, information relating to the bearer service is mapped into the BC (bearer capability) information element in GSM. Further details of the HLC, BC and also the LLC (low level capability element) are also provided in the following description of the mapping.

The V.8, V.8bis and GSM signalling information, which is relevant to the determination of telecommunication service types, is outlined below. The references refer to the following documents, the contents of which are incorporated herein by reference:

[1] ITU-T Recommendation F.721 (08/92) - Videotelephony teleservice for ISDN

[2] ITU-T Recommendation F.700 (07/96) - Framework Recommendation for audiovisual/multimedia services

[3] ITU-T Recommendation V.8 (02/98) - Procedures for starting sessions of data transmission over the public switched telephone network

[4] ITU-T Recommendation V.8bis (08/96) - Procedures for the identification and selection of common modes of operation between data circuit-terminating equipment (DCEs) and between data terminal equipment (DTEs) over the general switched telephone network and on leased point-to-point telephone-type circuits

[5] ITU-T Recommendation H.324 (2/98) - Terminal for low bit rate multimedia communication

[6] Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 specification, GSM 04.08 version 6.2.0 Release 1997

[7] ITU-T Recommendation H.245 (2/98) - Control protocol for multimedia communication.

V.8 signalling to be mapped [3]

Table 2 in [3] lists the information categories that are available in V.8 signalling. As can be seen, the call-function octet has 3 option bits, and Table 3 in [3] illustrates how these option bits are used to identify particular call functions. As mentioned above, the information interpreted from the call function category can be mapped into the Q.931 HLC information element to provide the GSM network with information pertaining to the telecommunications service or call function and the bearer service.

Table 4 in [3] indicates the availability of PSTN V-Series modulation modes other than V.90. The availability is only shown if the modulation mode can be used with the indicated call function, and if it is desired to convey that capability to the remote DCE. For example, if an H.324 application is used, the use of a V.34 full duplex mode is mandatory, and the other options would not be indicated. The modulation mode information is only used between DCEs in order to find out the common modulation modes. After selection of a common modulation mode (e.g. V.34), the MSC/IWF determines the data rate from the modem (e.g. 28.8kbps) on the basis of how good the quality (BER) of the PSTN line is.

GSM (Q931-based) signalling onto which the above V8 signalling is mapped by the MSC

GSM specification 04-08 [6] defines the messages for circuit switched call control. In particular, section 9.3.23.1 in [6] defines the SETUP message content for mobile terminated call establishment. This message is sent over the network to the mobile station to initiate mobile terminated call establishment. The SETUP message contains relevant information elements for the BC (bearer capability), HLC (high level capability) and LLC (low level capability). As mentioned above, it is these information elements which the V.8 call function category information is mapped into.

BC

The purpose of the bearer capability information element is to describe the bearer service to be used in the connection. Figure 10.5.88 in [6] illustrates the BC element for GSM. This element contains 16 octets (8 bit units). Certain bits or coding points need to be stored in octets 3, 4, 6, 6a, 6c and 6d in order to support H.324, as outlined below.

Transfer mode (octet 3):

Bit

4

0 **circuit mode**

1 packet mode

- Circuit mode is selected

Duplex mode (octet 4)

Bit

4

0 half duplex

1 **full duplex**

- Full duplex is selected

Synchronous/asynchronous (octet 6)

Bit

1

0 **synchronous**

1 asynchronous

- Synchronous is selected

User rate (octet 6a)

Bits

4 3 2 1	
0 0 0 1	0.3 kbit/s Recommendation X.1 and V.110
0 0 1 0	1.2 kbit/s Recommendation X.1 and V.110
0 0 1 1	2.4 kbit/s Recommendation X.1 and V.110
0 1 0 0	4.8 kbit/s Recommendation X.1 and V.110
0 1 0 1	9.6 kbit/s Recommendation X.1 and V.110
0 1 1 0	12.0 kbit/s transparent (non compliance with X.1 and V.110)
0 1 1 1	1.2 kbit/s/75 bit/s Recommendation V.23, (asymmetric) X.1, V.110.

The coding points shown above are used to indicate the user rate. At present there is no e.g. 28.8 kbps user rate available in GSM. However, Q.931 offers a wider set of user data rates that can be used when specifying the BC information element. (An outline of how the mapping might occur for the 28.8kbps and other unspecified data rates in GSM is outlined under the heading "Potential BC for other user rates").

Connection element (octet 6c)

Bit

7 6	
0 0	**transparent**
0 1	non transparent (RLP)
1 0	both, transparent preferred
1 1	both, non transparent preferred

- Transparent is selected

Other modem type (octet 6d)

Bits

7 6

0 0	no other modem type specified in this field
0 1	V.32bis
1 0	**V.34**

- V.34 is selected

<u>Potential BC for other bit user rates.</u>

Fixed network user rate (octet 6d)

Bit

5 4 3 2 1

0 0 0 0 0	Fixed network user rate not applicable/No meaning is associated with this value.
0 0 0 0 1	9.6 kbit/s Recommendation X.1 and V.110
0 0 0 1 0	14.4 kbit/s Recommendation X.1 and V.110
0 0 0 1 1	19.2 kbit/s Recommendation X.1 and V.110
0 0 1 0 0	**28.8 kbit/s** Recommendation X.1 and V.110
0 0 1 0 1	38.4 kbit/s Recommendation X.1 and V.110
0 0 1 1 0	48.0 kbit/s Recommendation X.1 and V.110(synch)
0 0 1 1 1	56.0 kbit/s RecommendationX.1 and V.110(synch) /bit transparent
0 1 0 0 0	64.0 kbit/s bit transparent

For an example data rate of 28.8 kbps, the element highlighted above might be used. So 28.8 is selected.

Acceptable channel codings (octet 6e), mobile station to network direction:

Bit

7

0 TCH/F14.4 not acceptable

1 TCH/F14.4 acceptable

If 14.4 is implemented then it can be selected.

Bit

6

0 Spare

Bit

5

0 TCH/F9.6 not acceptable

1 TCH/F9.6 acceptable

Normally, 9.6 is implemented as well

Bit

4

0 TCH/F4.8 not acceptable

1 TCH/F4.8 acceptable

4.8 exists even though not widely used

Acceptable channel codings (octet 6e), network to MS direction:

Bits 4 to 7 are spare and shall be set to "0".

This would mean that network cannot decide what kind of channel coding is used. This is required in the present case since networks initiate the call setup renegotiation. A change to current GSM implementation is needed to effect this.

Maximum number of traffic channels (octet 6e), MS to network direction:

Bits

3 2 1

0 0 0 1 TCH

0 0 1 2 TCH

0 1 0   3 TCH
0 1 1   4 TCH
1 0 0   5 TCH
1 0 1   6 TCH
1 1 0   7 TCH
1 1 1   8 TCH

28.8kbps can be obtained by combining two 14.4 channels using HSCSD. This could also be done by combining three 9.6 channels.

Maximum number of traffic channels (octet 6e), network to MS direction:
Bits 1 to 3 are spare and shall be set to "0".
As above, this would mean that network is unable to decide what kind of channel coding is used, and would require a modification to GSM.

BC Information derived from Q931

BC information used in GSM in-band signalling is derived from Q.931 and modified for the purposes of GSM. The BC element discussed above is proposed to be used in UMTS in a frame of reference of this patent application.

In the Q.931 BC information element there exists one additional coding point which is needed and should be included in UMTS signalling. This coding point is User information layer 1 protocol (octet 5) and is described below:

User information layer 1 protocol (octet 5)
Bits

5 4 3 2 1
00001   CCITT standardised rate adaption V.110, I.460 and X.30. This implies the presence of octet 5a and optionally octets 5b, 5c and 5d as defined below.

00010	Recommendation G.711 μ-law
00011	Recommendation G.711 A-law
00100	Recommendation G.721 32 kbit/s ADPCM and Recommendation I.460.
00101	Recommendations H.221 and H.242
00110	**Recommendations H.223 and H.245**
00111	Non-ITU-T standardized rate adaption. This implies the presence of octet 5a and, optionally, octets 5b, 5c and 5d. The use of this code point indicates that the user rate specified in octet 5a is defined by the user. Additionally, octets 5b, 5c and 5d, if present, are defined consistent with the user specified rate adaption.
01000	ITU-T standardised rate adaption V.120. This implies the presence of octets 5a and 5b as defined below, and optionally octets 5c and 5d.
01001	CCITT standardised rate adaption X.31 HDLC flag stuffing.
	All other values are reserved.

- Recommendations H.223 and H.245 is selected

LLC

The low layer compatibility information element is illustrated in Figure10.5.104 of [6]. The purpose of the low layer compatibility information element is to provide a means which should be used for compatibility checking by an addressed entity (e.g., a remote user or an interworking unit or a high layer function network node addressed by the calling user). The Low Layer Compatibility information element is transferred transparently by a PLMN between the call originating entity (e.g. the calling user) and the addressed entity.

Except for the information element identifier, the low layer compatibility information element is coded as in ETS 300 102-1.

This information element is not necessarily needed because the same coding can be conveyed in the BC.

HLC

The purpose of the high layer capability information element is to provide a means by which the remote user can check for compatibility.

The high layer compatibility information element is coded as shown in figure 10.5.102 and table 10.5.125 of [6].

The High Layer Compatibility information element is transported transparently by a PLMN between a call originating entity (e.g. a calling user) and the addressed entity (e.g. a remote user or a high layer function network node addressed by the call originating entity). However, if explicitly requested by the user (at subscription time), a network which provides some capabilities to realise teleservices may interpret this information to provide a particular service.

The following HLC code points can be found from the Q.931 recommendation:

High layer characteristics identification (octet 4)
Bits
7 6 5 4 3 2 1
0 0 0 0 0 0 1Telephony
0 0 0 0 1 0 0 Facsimile Group 2/3 (Recommendation F.182 [68])
0 1 0 0 0 0 1 Facsimile Group 4 Class I (Recommendation F.184 [69])

0 1 0 0 1 0 0 Teletex service, basic and mixed mode of operation (Recommendation F.230 [70]) and facsimile service Group 4, Classes II and III (Recommendation F.184)

0 1 0 1 0 0 0 Teletex service, basic and processable mode of operation (Recommendation F.220 [71])

0 1 1 0 0 0 1 Teletex service, basic mode of operation (Recommendation F.200 [72])

0 1 1 0 0 1 0 Syntax based Videotex (Recommendations F.300 [73] and T.102 [74])

0 1 1 0 0 1 1 International Videotex interworking via gateways or interworking units (Recommendations F.300 and T.101 [75])

0 1 1 0 1 0 1 Telex service (Recommendation F.60 [76])

0 1 1 1 0 0 0 Message Handling Systems (MHS) (X.400 - Series Recommendations [77])

1 0 0 0 0 0 1 OSI application (Note 2) (X.200 - Series Recommendations [78])

1 0 1 1 1 1 0 Reserved for maintenance (Note 4)

1 0 1 1 1 1 1 Reserved for management (Note 4)

1 1 0 0 0 0 0 Audio visual (Recommendation F.721 [79])

1 1 0 0 0 0 1 through 1 1 0 1 1 1 1 Reserved for audiovisual services [2]

1 1 1 1 1 1 1 Reserved

The F.700 Recommendation [2] provides a methodology for constructing multimedia services which is timely and responsive to the needs of both the End-User and Service Provider. This methodology links generic End-User application requirements with the established set of network independent building blocks or other communications capabilities being developed within ITU-T. The procedures described in this Recommendation are intended for use in developing a series of General Service Recommendations which combine the required communication tasks and media components into an architecture for generic services (e.g. for Multimedia Conferencing Service, Multimedia Distribution Service, etc.). Where applicable Recommendations are not yet available, this methodology can be used as the basis for a

structured dialogue between End-Users and Service Providers in arriving at a responsive service solution.

Bits 1100001 through 1101111, specified in F.700, could be used for e.g. H.324 code point or a totally new codepoint could be designed for UMTS use.

V.8bis signalling to be mapped [4]

Table 6-2 in [4] lists the standard information categories that are available in V.8bis signalling. It comprises categories such as data and H.324 multimedia terminal to readily cater for such future teleservices. The coding for these teleservices are illustrated in Tables 6-3 and 6-5 of [4]. One advantage of the V.8bis capability exchange is that, in most cases, it enables terminals to ascertain very quickly when operation in H.324 mode is desired. This is because V.8bis capabilities indicate only the most basic and commonly used modes. If an H.324 operation mode not signalled by V.8bis is desired, the terminal must complete call establishment and perform a H.245 [7] capabilities exchange to determine if the far-end terminal supports the desired mode.

Within the Rec. V.8bis Communications Capabilities (CC) field for H.324, the CC field is formatted into one or more sub-fields. Each sub-field ends with the octet in which bit [n] is set to 1. Following the first sub-field, the remaining sub-fields, if present, shall appear in the same order in which the bits indicating their presence are transmitted. The actual bit assignments can be seen from [4].
In the first sub-field the following bits are allocated:

Name	Meaning
Video	Shall be set only if bi-directional video is supported per Rec. H.324 (sec. 6.6).

Audio — Shall be set only if bi-directional audio is supported per Rec. H.324 (sec. 6.7).

Encryption — Shall be set only if encryption is supported per Rec. H.324 (sec. 9.2).

Data — Indicates that a data subfield is present. Shall be set only if one or more bits in the data subfield are set.

Possible future allocations include Profiles (new subfield).

In the Data subfield, the following bits are allocated:

Name Meaning

T.120 Shall be set only if T.120 conferencing is supported per Rec. H.324 (sec. 6.8.2.1).

T.84 Shall be set only if T.84 still image transfer is supported per Rec. H.324 (sec. 6.8.2.2).

T.434 Shall be set only if T.434 file transfer is supported per Rec. H.324 (sec. 6.8.2.3).

V.42 Shall be set only if V.42 user data is supported per Rec. H.324 (sec. 6.8.1.2/6.8.2.6).

V.14 Shall be set only if V.14 user data is supported per Rec. H.324 (sec. 6.8.1.1/6.8.2.6).

PPP Shall be set only if IETF Point-to-Point protocol is supported via the Network Layer Protocol Identifier (NLPID) per Rec. H.324 (sec. 6.8.2.5).

T.140 Shall be set only if T.140 Text Conversation Protocol for Multimedia Application is supported per Rec H.324 (sec 6.8.2.8).

Other modes beside those indicated in V.8bis, such as unidirectional modes, may be supported by terminals as signalled via H.245 capabilities exchange.

With V.8bis signalling can also be used when the call is first started in speech mode and after that it is switched to e.g. H.324 mode. For this reason, the MSC/IFW must all the time be able to listen the possible inband signalling coming from PSTN modem. The switch of service then initiates the bearer renegotiation in UMTS side where the "old" bearer is accomplished according to QoS parameters needed to H.324 call.

Mapping between UMTS and PSTN/ISDN

In this section different information elements and their codings are described. These codings must be mapped in the interface of either PSTN / UMTS or ISDN / UMTS.

Necessary PSTN codings in the frame of reference of the patent application:

V.8

	Call function category (Octet – "callf0")
100	PSTN Multimedia Terminal (Bits 567)
	Modulation modes category (Octet – "modn0")
1	V.34 duplex availability (Bit 6)

V.25ter

	Modulation control commands
	Modulation reporting control (+MR)
+MCR: V34	+MCR: <carrier>
+MRR: 28800	+MRR: <rate>

V.8bis

Standard information field - {SPar(1)} coding

Standard information field – Data {NPar(2)} coding (Octet 2)

1 Rec. V.34 duplex mode (Bit 5)

Standard information field - {SPar(1)} coding

Standard information field – H.324 multimedia terminal {NPar(2)} coding

1 Video (Bit 1)

Standard information field - {SPar(1)} coding

Standard information field – H.324 multimedia terminal {NPar(2)} coding

1 Audio (Bit 2)

Necessary ISDN codings (in a case V.140 signalling is needed) in the frame of reference of the patent application:

V.140 inband signals consist of HDLC-framed PDUs defined using ASN.1 syntax according to Recommendation X.680 and coded according to the packed encoding rules of Recommendation X.691.

The purpose why V.140 signalling is used is that there exist networks that fail to send out band signalling messages end to end. V.140 signalling used in ISDN does more or less the same thing as V.8/V.8bis signalling in PSTN.

The capabilitySet field of the roleAndCapability PDU contains a sequence of one or more Capability structures, each expressing the terminal's ability to work in a particular multimedia or other communication protocol. The transmitter shall include the complete list of modes in which it is currently able to operate. The list of possible modes is defined in Annex A of V.140, and

may be extended in the future. Capabilities shall be listed in order of preference, from most-preferred to least-preferred. In our case the h324AnnexD should be the first (or only one) of the listed capabilities in capabilitySet field.

The receiving terminal can answer the transmitting terminal of the selected mode by using modeSelect PDU.

In the following the syntax of PDUs using ASN.1 notation is described [V.140 Annex A]. The "code point" of our choice is presented with bigger font.

```
HDISPATCH DEFINITIONS AUTOMATIC TAGS ::=
BEGIN

-- Export all symbols

--
==============================================================
-- Top level PDUs
--
==============================================================

HDispatchPDU                      ::=CHOICE
{
    nonStandard                   NonStandardMessage,
    roleAndCapability             RoleAndCapabilityMessage,
    modeSelect                    Mode,
    youChoose                     NULL,
    modeSelectAcknowledge         NULL,
    terminate                     TerminateMessage,
    ...
}

RoleAndCapabilityMessage          ::=SEQUENCE
{
    role                          CHOICE
    {
        originate                 NULL,
        unknown                   NULL,
        answer                    NULL,
        ...
    },
```

```
    arbitrationField              INTEGER (0..4294967295), -- 32 bit
                                  random #
    capabilitySet                 SEQUENCE SIZE (1..65535) OF
                                  Capability,
    ...
}

Capability                        ::= CHOICE
{
    nonStandard                   NonStandardParameter,
    isdn                          CHOICE
    {
    isdnCapability                IsdnCapability,
    multilinkAdditionalConnection NULL,   -- Express this cap alone to
                                          force association of this
                                          channel with
                                                --an existing call
    is13871                       SEQUENCE      -"BONDING" protocol
        {
                                  withIsdnCapability    IsdnCapability,
                                  ...
        },

        h244              SEQUENCE                 --        channel
aggregation protocol
        {
                          withIsdnCapability       IsdnCapability,
                          ...
        },
        ...
    },
    ...
}

IsdnCapability                  ::=CHOICE
{
    g711aLaw            SEQUENCE {...},
    g711uLaw            SEQUENCE {...},
    h320                SEQUENCE {...},
    h324AnnexD          SEQUENCE {...},
    h324Multilink       SEQUENCE {...},
    group4Fax           SEQUENCE {...},
    t120                SEQUENCE {...},
    t140                SEQUENCE {...},     -- text chatting protocol
    v110                SEQUENCE {...},
    v120                SEQUENCE {...},
    rfc1661             SEQUENCE
    {
        withH323        BOOLEAN,
```

```
        ...
    },
    ...
}


{
    nonStandard                         NonStandardParameter,
    plainIsdnMode                       IsdnMode,
    h244                                IsdnMode,
    is13871                             IsdnMode,        --      BONDING
protocol
    multilinkAdditionalConnection       SEQUENCE
    {
        callAssociationNumber       INTEGER (0..4294967295),
        ...
    },
    ...
}

IsdnMode                        ::= CHOICE
{
    nonStandard                     NonStandardParameter,
    g711aLaw                        SEQUENCE {...},
    g711uLaw                        SEQUENCE {...},
    h320                            SEQUENCE {...},
    h324AnnexD                      SEQUENCE {...},
    h324Multilink                   SEQUENCE {...},
    group4Fax                       SEQUENCE {...},
    t120                            SEQUENCE {...},
    rfc1661                         SEQUENCE {...},
    ...
}


TerminateMessage                    ::=SEQUENCE
{
    cause                           CHOICE
    {
        nonStandard                 NonStandardParameter,
        timerExpiration             NULL,
        roleCollision               NULL,
        noSuitableModes             NULL,
        invalidModeSelected         NULL,
        protocolViolation           NULL,
        modeNotAvailable            NULL,
        ...
    },
```

```
    ...
}

--
===========================================================
-- Non standard Message definitions
--
===========================================================

NonStandardMessage              ::=SEQUENCE
{
    nonStandardData             NonStandardParameter,
    ...
}

NonStandardParameter            ::=SEQUENCE
{
    nonStandardIdentifier       NonStandardIdentifier,
    data                        OCTET STRING
}

NonStandardIdentifier           ::=CHOICE
{
    object                      OBJECT IDENTIFIER,
    h221NonStandard             SEQUENCE
    {
        t35CountryCode          INTEGER (0..255),-- country, per T.35
        t35Extension            INTEGER (0..255),-- assigned ationally
        manufacturerCode        INTEGER (0..65535)-- assigned
nationally
    }
}
```

Necessary UMTS (GSM) codings in the frame of reference of the patent application:

<u>Bearer Capability</u>

	Bearer Capability Information Element
	Transfer mode (octet 3)
0	Circuit mode (Bit 4)
	Bearer Capability Information Element
	Duplex mode (octet 4)
1	Full duplex (Bit 4)
	Bearer Capability Information Element

	Synchronous/Asynchronous (octet 6)
0	Synchronous (Bit 1)
	Bearer Capability Information Element
	Fixed network user rate (octet 6d)
00100	28.8 kbps (Bits 54321)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F14.4 acceptable (Bit 7)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F9.6 acceptable (Bit 5)
	Bearer Capability Information Element
	Acceptable channel codings (octet 6e)
1	TCH/F4.8 acceptable (Bit 4)
	Bearer Capability Information Element
	Maximum number of traffic channels (octet 6e)
001	2 TCH (Bits 321)
	Bearer Capability Information Element
	Connection element (octet 6c)
00	Transparent (Bits 76)
	Bearer Capability Information Element
	Other modem type (octet 6d)
10	V.34 (Bits 76)
	Bearer Capability Information Element
	User information layer 1 protocol (octet 5)
00110	Recommendations H.223 and H.245 (Bits 54321)

The latest information coding point is not present in current GSM BC information element. It is taken from Q.931 and should be added to future UMTS BC information element.

<u>High Layer Compatibility</u>

	High Layer Compatibility Information Element
	High layer characteristics identification (octet 4)
1100001	PSTN Multimedia Terminal (Bits 7654321)

The present invention includes any novel feature or combination of features disclosed herein either explicitly or any generalisation thereof irrespective of whether or not it relates to the claimed invention or mitigates any or all of the problems addressed.

In view of the foregoing description it will be evident to a person skilled in the art that various modifications may be made within the scope of the invention. For example, the reduced numbering scheme is not restriceted to only one or two numbers for call type differentiation. It will be appreciated that the reference to H.324 throughout this text is purely exemplary and that the invention is also applicable to other data types.